REMARKS

These Remarks are in reply to the final Office Action mailed July 23, 2010. Claims 1, 2, 7-12 and 19-29 were pending in the Application prior to the outstanding Office Action. No claims are currently being amended, canceled or added. Accordingly, claims 1-2, 7-12 and 19-29 remain pending for the Examiner's consideration, with claims 1, 10 and 22 being independent. In view of the following remarks, Applicants respectfully request that the outstanding rejections be reconsidered and withdrawn.

I. Summary of Prior Art Claim Rejections

Claims 1, 2, 7, 10-12, 19 and 22-29 were rejected under 35 U.S.C. §103(a) as allegedly being unpatentable over U.S. Patent No. 4,841,828 to Suzuki (hereafter referred to as "Suzuki") in view of U.S. Patent No. 5,023,825 to Luthra et al. (hereafter referred to as "Luthra"), U.S. Patent No. 5,928,313 to Thompson (hereafter referred to as "Thompson") U.S. Patent No. 5,471,411 to Adams et al. (hereafter referred to as "Adams") and U.S. Patent No. 4,727,505 to Konishi et al. (hereafter referred to as "Konishi").

Claims 8, 9, 20 and 21 were rejected under 35 U.S.C. §103(a) as allegedly being unpatentable over Suzuki in view of Luthra, Adams, Thompson, Konishi and further in view of U.S. Patent No. 6,411,333 to Auld et al. (hereafter referred to as "Auld").

(a Discussion of the Claims begins on the next page)

II. Discussion of the Claims

Claim 1 is reproduced below for the convenience of the Examiner.

1. (Previously Presented) A method comprising:

(a) storing a plurality of independent sets of filter coefficients in a memory, wherein each set of filter coefficients defines a different polyphase filter function, wherein each of the different polyphase filter functions would result in at least some modifying of a signal if the signal were filtered in accordance with the polyphase filter function, and wherein each of the different polyphase filter functions would result in modifying of a signal in a different manner than the other polyphase filter functions;

(b) storing a value in a filter selection register;

(c) selecting a single one of the independent sets of filter coefficients based on the value stored in the filter selection register;

(d) receiving an audio input signal including a plurality of samples;

(e) estimating a sample rate of the audio input signal;

(f) interpolating the single one selected set of filter coefficients, in dependence on the estimated sample rate of the audio input signal, to thereby produce interpolated polyphase filter coefficients; and

(g) convolving the produced interpolated polyphase filter coefficients with the samples of the audio input signal to produce a filtered audio output signal that differs from the audio input signal regardless of which single one of the sets of filter coefficients is selected;

wherein said selecting the single one of the independent sets of filter coefficients at step (c) is performed prior to receiving the audio input signal at step (d), independent of the audio input signal received at step (d), and independent of the filtered audio output signal produced at step (g); and

wherein the same single one of the sets of filter coefficients selected at step (c) is used at steps (f) and (g) to produce the filtered audio output signal produced at step (g).

As shown above, claim 1 requires the following features:

> "wherein said selecting the single one of the independent sets of filter coefficients at step (c) is performed prior to receiving the audio input audio signal at step (d), independent of the audio input signal received at step (d), and independent of the filtered audio output signal produced at step (g); and
>
> wherein the same single one of the sets of filter coefficients selected at step (c) is used at steps (f) and (g) to produce the filtered audio output signal produced at step (g)."

In the previous Reply filed on May 11, 2010, Applicants explained in great detail why Thompson (which had been relied upon in the previous Office Action to allegedly teach the features of then dependent claim 5) did not teach or suggest these features of claim 1.

In the outstanding Office Action, it is conceded that Suzuki does not teach these features of claim 1. However, in the Response to Arguments on pages 2 and 3 of the Office Action, it appears to have been asserted that it is Luthra that is being relied upon to disclose these features of claim 1, and thus, it appears that the Examiner agrees that Thompson does not disclose these features of claim 1. With regards to Luthra, at item 6 on page 4 of the Office Action it is asserted that the Abstract and column 2, lines 9-28 of Luthra disclose these features because these portions of Luthra "disclose a filter obtaining an output sample rate by utilizing an appropriate set of coefficients, from a plurality of sets, from a particular bin; thus selecting an independent set of filter coefficients ... from the bin independently of the input and output". For at least the reasons set forth below, Applicants respectfully disagree with this assertion.

The Abstract and column 2, lines 9-28 of Luthra are reproduced below for the convenience of the Examiner.

> ABSTRACT
>
> A technique for reducing the number of coefficients in a low ratio sampling rate converter divides each input sampling rate interval into a plurality of bins. For each bin a set of conversion coefficients is computed. So long as output data samples lie within the same bin, the same set of coefficients is applied. When the output data samples move to another bin, the set of coefficients applied is changed. Thus a finite set of coefficients are computed rather than computing an exact set for each output data sample.
> *(Abstract of Luthra)*

> SUMMARY OF THE INVENTION
>
> Accordingly the present invention provides a technique for reducing the number of coefficients for a filter in a low ratio sampling rate converter by dividing the sampling interval of the input sampling rate into a finite number of bins and precalculating the coefficients for the bins to produce the output sampling rate. A finite input response (FIR) filter receives at its input at the output sampling rate a plurality of samples that were obtained at the input sampling rate. A coefficient memory contains the coefficients for each bin. When the output samples move to the next bin, a coefficient controller accesses the coefficient memory to output a new set of coefficients for the FIR filter. Since there are multiple samples per bin when the ratio is very nearly one, a slow, inexpensive memory may be used as the coefficient memory. Also for on the fly operation the coefficients may be calculated at a rate significantly less than the video rates, eliminating the need for the coefficient memory altogether.
> *(column 2, lines 9-28 of Luthra)*

The Abstract of Luthra explains that for each of a plurality of input sample rate bins, a set of filter coefficients is computed. The Abstract further states that "[s]o long as output data samples lie within the same bin, the same set of coefficients is applied", but that "[w]hen the output data samples move to another bin, the set of coefficients applied is changed." Thus, the Abstract of Luthra makes it very clear that the applied set of coefficients **<u>is dependent</u>** on the output data samples, and that the applied set of coefficients **<u>changes</u>** based on the output data samples. Thus, Luthra clearly does not teach or suggest that "said selecting ***the single one*** of the independent sets of filter coefficients at step (c) is performed prior to receiving the audio input audio signal at step

(d), ***independent*** of the audio input signal received at step (d), and ***independent*** of the filtered audio output signal produced at step (g); and wherein the ***same single one of the sets*** of filter coefficients selected at step (c) is used at steps (f) and (g) to produce the filtered audio output signal produced at step (g)", as required by claim 1.

Further, column 2, lines 9-18 of Luthra states that "[w]hen the output samples move to the next bin, a coefficient controller accesses the coefficient memory to output a ***new set of coefficients*** for the FIR filter." Further, column 2, lines 9-18 of Luthra also states that "for on the fly operation the coefficients may be calculated at a rate significantly less than the video rates, eliminating the need for the coefficient memory altogether". This portion of Luthra also makes it very clear that in Luthra filter coefficients are dependent on output data samples (and thus the output signal), and that in Luthra filter coefficients change based on the output data samples (and thus the output signal). This again supports Applicants assertion that Luthra clearly does not teach or suggest that "said selecting ***the single one*** of the independent sets of filter coefficients at step (c) is performed prior to receiving the audio input audio signal at step (d), ***independent*** of the audio input signal received at step (d), and ***independent*** of the filtered audio output signal produced at step (g); and wherein the ***same single one of the sets*** of filter coefficients selected at step (c) is used at steps (f) and (g) to produce the filtered audio output signal produced at step (g)", as required by claim 1.

As mentioned above, it was already conceded in the Office Action that Suzuki does not teach these features of claim 1. In the previous Reply filed on May 11, 2010, Applicants explained in great detail why Thompson did not teach or suggest these features, and as mentioned above, it appears that the Examiner agrees that Thompson does not disclose these features of claim 1. Applicants have just explained above why Luthra does not teach or suggest these features of claim 1. Further, Adams and Konishi, alone or in combination, do not teach or suggest these features of claim 1. For example, Adams explains that its "coefficient address generator 112 ... generates the locations of coefficients corresponding to the accessed input data values" (see Adams, column 7, lines 14-18), which implies that Adams selects coefficients based on an input signal. Further, because Konishi does **not** relate to storing a plurality of independent sets of filter

coefficients, and does **not** select one of the independent sets of filter coefficients, it would not make any sense for Konishi to teach the above discussed features of claim 1. Rather, Konishi appears to store only a single set of 12 coefficient data factors g(k), which is half of the symmetric 24 coefficient data factors that Konishi says are necessary for one convolution processing cycle in digital filtering (see Konishi, column 6, lines 58-62). While it may be argued that Konishi uses these same stored 12 coefficient data factors g(k) independent of the input signal and the output signal, this is simply because Konishi never selects one set of filter coefficients from a plurality of independent sets of filter coefficients. In other words, where only one set is stored (as in Konishi), the same set inevitably will be used at all times. Stated still another way, Konishi's using one set of coefficients (because only one set of coefficients is available) can not possible teach or suggest selecting one of many different sets prior to receiving an audio input signal, ***independent*** of the received audio input signal and ***independent*** of a produced filtered audio output signal, as required by claim 1.

For at least the reasons specified above, Applicants respectfully request that the rejection of claim 1 be reconsidered and withdrawn.

Claims 2, 7-9, 24 and 27 depend from and add additional features to claim 1. Applicants respectfully assert that these claims are patentable for at least the reason that they depend from claim 1, as well as for the features that they add.

Claim 10 is believed to be patentable over the cited references for similar reasons to at least some of the reasons discussed above with regards to claim 1. Accordingly, Applicants respectfully request that the rejection of claim 10 be reconsidered and withdrawn. **Claims 11, 12, 19-21, 25 and 28** depend from and add additional features to claim 10. Applicants respectfully assert that these claims are patentable for at least the reason that they depend from claim 10, as well as for the features that they add.

Claim 22 is believed to be patentable over the cited references for similar reasons to at least some of the reasons discussed above with regards to claim 1. Accordingly, Applicants respectfully request that the rejection of claim 22 be reconsidered and

withdrawn. **Claims 23, 26 and 29** depend from and add additional features to claim 22. Applicants respectfully assert that these claims are patentable for at least the reason that they depend from claim 22, as well as for the features that they add.

III. Conclusion

In light of the above, it is respectfully requested that all outstanding rejections be reconsidered and withdrawn. The Examiner is respectfully requested to telephone the undersigned if he can assist in any way in expediting issuance of a patent.

The Commissioner is authorized to charge the required fees and any underpayment of fees or credit any overpayment to Deposit Account No. 06-1325 for any matter in connection with this reply, including any fee for extension of time, which may be required.

Respectfully submitted,

Date: September 22, 2010

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